REMARKS

1. Responsive to the Examiner's objections to the claims, the language of claim 29 has been changed to read "a number" rather than "the number". The word "temporally" that appears in claim 8 is not an error, but rather serves to indicate that the stored template is temporally (in the time domain) associated with the signal. The Examiner's substantive rejections of the claims based upon cited prior art references are respectfully traversed herein, and reconsideration is respectfully requested.

2. The Examiner has rejected claims 1, 4-8, 10, 12-14, 23, 25-28 and 34 under 35 U.S.C. § 102(e) as being anticipated by Chan et al. (U.S. Patent No. 6,870,807). Chan et al. is directed toward suppressing music on hold in a conference call environment where the conference call is managed by a digital PBX. In the embodiment illustrated and described in Chan et al., a conference bridge is established within the digital PBX that simply directs the conference call audio time slots to the appropriate listeners. For example, for user A (in FIG. 1 of Chan et al.), whose audio stream is in time slot TS#1, the digital PBX directs user A to listen to time slots TS#2, TS#3, and TS#4, which correspond to the audio streams of the other conference call participants.

In the event that music is detected in one of the conference call time slots (or one of the users enters a suppression command), the offending time slot is simply omitted from the conference call summation until the system determines, through periodic monitoring, for example, that the music has ceased.

The invention described and claimed by the applicants instead relates generally to assisting a calling party whose call has been queued; for example, by a call center. When a party

calls a customer service number for assistance, the call may be routed to a call center where the call is queued until a customer service representative is available to take the call. Frequently, music and/or repetitive messages are played until the call is finally answered by customer service.

The caller is rescued from this music and endless stream of voice messages by the audio stream monitor service of the present invention. The audio stream monitoring service is provided by the telephone network and can be activated for calls placed into a call queue. The audio stream monitor service removes repetitive unwanted messages from the audio stream. In addition, the monitor service provides the caller with estimates of how long the call will remain queued, either via audio on the telephone line or via display, such as through TV or a personal computer.

One should keep in mind that in a telephone network, there are at least two kinds of intelligent network devices. First, there is an SCP, or service control point, which has only a signaling connection to switches in the telephone network (no voice, in other words). The SCP's role is to provide service logic and database access for switches.

Another type of intelligent network device is the service node, or SN. The SN has both bearer and signaling connections to the switches in the network. In one embodiment of this invention, it is the SN that filters the voice (audio) stream that passes through it for queued calls. In another embodiment, the SN relays updates from the call center (an audio stream source) to the caller regarding estimated remaining wait time. The SN is referred to as an intelligent network node, or IN node, in the drawing figures.

When a caller is connected to a call center, the SCP recognizes this (possible through connection to an external database), and further recognizes that the caller is a subscriber to the ASMS (audio stream monitor service) located on the intelligent network node.

The SCP then orders the switch to transfer the call path so that the caller is connected to the SN (the intelligent network node). The SCP also order the SN to offer the ASMS, identifies the telephone number of the call center, and supplies other caller related information it may have gathered from it database queries, for example, information on caller devices capable of receiving audio and text.

The SN then calls that telephone number (the call center, in other words), and, upon answer, begins monitoring the call stream in both directions between the call center and the caller. The ASMS executing on the SN removes selected audio segments for the audio stream flowing from the call center. Often-repeated messages, such as "Your call is important to us . . ." and its many variations, can be removed in at least two ways. One way is through comparison with a database of audio stream segments, digitally encoded, compiled by the service provider and keyed with the telephone number dialed by the caller when the call center session is initiated.

Another way in which audio stream segments may be identified for removal is by receiving a signal from the caller during the segment, or shortly after permitting the segment to flow. The signal could be a touch-tone key sequence, for example. The ASMS would maintain a record of the flagged audio stream and remove subsequent instances.

In another embodiment of the invention, the user may call a telephone number assigned to the ASMS itself when the user suspects that a call center will be involved in his call. The

ASMS then prompts the user to dial the call center number, and this keeps the ASMS involved in the call from the very beginning, filtering out unwanted audio segments.

In yet another embodiment, the user calls the call center directly. After the call is established, and after the call is queued, the user keys in a particular sequence (for example), or uses a hookswitch or flash. The switch recognizes this signal and queries the SCP for direction. The SCP causes the switch to direct the audio stream through the SN, and the SN activates the ASMS.

In another embodiment, there are cooperating call centers that provide an estimate of how long it will be until a service representative is available to take the call. The intelligent network node translates wait time information into a suitable format for transmission to the caller. The wait time information may be sent by audio to the calling telephone set, or it may be sent to an associated television set or PC.

The Chan et al. reference simply fails to meet the limitations of the applicants' independent claims 1 and 23, in that the digital PBX is not an intelligent network node. Furthermore, the digital PBX is not able to filter a portion of the audio stream and deliver a filtered audio stream to a call participant. In Chan et al., the offending time slot is simply disconnected from the call. Also, Chan et al. can detect music, but cannot detect other types of potentially offending message streams, and, consequently, fails to disclose each and every element of the applicants' invention. Consequently, the applicants respectfully submit that claims 1 and 23 are allowable over Chan et al. Claims 4-8, 10, and 12-14, which depend ultimately from claim 1 are also allowable over Chan et al. as depending from an allowable base claim. The same is true of claims 25-28 and 34, which depend ultimately from claim 23.

3. The Examiner has rejected claim 9 under 35 U.S.C. § 103(a) as unpatentable over Chan et al. in view of Light et al. (U.S. Patent No. 6,349,136). The Light et al. reference is directed toward a system for controlling a conference call. In the system of Light et al., a telephone switching system (20 in FIG. 1 of Light et al.) includes a well-known conference bridge 75 for connecting the parties to a conference call. If one of the participants is placing music on hold on his line, or if one of the conference lines is excessively noisy, this condition is detected through personal monitoring by one of the conference participants, who is denominated the "controlling participant." By entering an appropriate control code via his telephone keypad, the controlling participant causes the conference participants to be connected through the conference bridge in sequential fashion. The controlling participant listens to each line in turn, and decides whether a given participant should be connected to or dropped from the conference call.

Claim 9 depends ultimately from claim 1, which has been shown to be allowable over Chan et al. above. In addition, the limitation introduced by claim 9, particularly when considered in light of the parent claims, cannot be said to be made obvious through consideration of Light et al. in conjunction with Chan et al. for the reasons set forth above. Consequently, the applicants respectfully submit that claim 9 is patentably distinguishable over Light et al. in combination with Chan et al.

4. The Examiner has rejected claim 11 under 35 U.S.C. § 103(a) as unpatentable over Chan et al. in view of Gopalakrishnan et al. (U.S. Patent No. 5,848,163). The Gopalakrishnan et al. reference is directed toward suppressing background music or noise from the input of a speech recognizer. Since additive background music or noise corrupts the speech

input that the system is designed to recognize, Gopalakrishnan et al. utilize an adaptive filter to selectively remove the unwanted noise signals (background music or noise) from the speech signals that the system is designed to recognize. The filtered speech signal is then applied to the speech recognizer.

Claim 11 depends ultimately from claim 1, which has been shown to be allowable above. In addition, contrary to the Examiner's assertion, any "gaps" that may occur in Gopalakrishnan et al. correspond to either music or noise, and are not gaps in the sense that the applicant has disclosed and claimed this concept. Consequently, claim 11 is also patentably distinguishable over the combination of Chan et al. and Gopalakrishnan et al., and is thus in condition for allowance for this additional reason.

5. The Examiner has rejected claim 33 under 35 U.S.C. § 103(a) as unpatentable over Chan et al. in view of Marks et al. (U.S. Patent No. 5,844,896). The Marks et al. reference is directed toward routing telephone calls with an AIN (Advanced Intelligent Network). Marks et al. describe a way in which the features of an AIN can be exploited to overcome a perceived problem with call-queuing systems. The problem identified by Marks et al. is that, when all lines within a Hunt group are busy, the call handling system will either generate a message stating that all lines are currently busy, and calls will be answered in the order received, or the call handling system will require the caller to leave a voice message so that his or her call can be returned by the called party.

In the view of Marks et al., the caller should be given the option of either remaining on the line or recording a message. In the system of Marks et al., shown in FIG. 2, a DMS-100 subscriber switch 30 communicates with an intelligent peripheral (IP) 42 through a service

control point (SCP) 32 to create an AIN. When a call to a queuing system cannot be completed, the call is forwarded to the IP. When the queuing system is able to accept the call, the call is simply placed by the IP. However, the IP is also responsive to caller inputs, and the caller may elect to disconnect or leave a message for forwarding to the queuing system when a line becomes available.

It is noteworthy that the Marks et al. reference is concerned with monitoring and generation of common channel signaling (SS7) in the unique environment of an AIN, and mentions bearer traffic only in passing. Marks et al. also requires that the system have TAT (Terminating Attempt Trigger) and terminating Next Event List (NEL) functionality in order to operate.

Applicants' claim 33 depends ultimately from claim 23, which has been shown to be allowable above. In addition, the essential limitations of claim 33 introduce patentable subject matter, particularly when considered in light of the parent claim. The context in which a service control point is mentioned in Marks et al. is completely unrelated to the system of Chan et al., as discussed above, so combination of these two references is inappropriate in an effort to yield the limitations of claim 33. Consequently, claim 33 is in condition for allowance for this additional reason.

6. The Examiner has rejected claims 29-32 under 35 U.S.C. § 103(a) as unpatentable over Chan et al. in view of Horn (U.S. Patent No. 6,556,670). The Horn reference is broadly directed toward a method for solving the "music-on-hold" problem in an audio conference. This reference concerns itself with the situation in which a conference call is taking place, and one of the participants places the call on hold. If the participant who places the call on hold has a

music-on-hold feature, then this music is effectively broadcast to all of the conference participants while the hold is in progress.

It is well-known in the art that a conference bridge has the capability to merge and/or sum selected audio signals originating from multiple conference call participants such that each participant can both hear and speak to all of the other participants. The embodiments described in Horn differ is several important particulars from the embodiment described in the present invention. First, a conference bridge has only a voice traffic connection to the switching devices (central office switches, network switches, and PBXes, for example) of the telecommunications network. In other words, a conference bridge can access particular voice traffic, but has no signaling connection to the switches in the network. Consequently, a conference bridge cannot perform the origination functions described and claimed by the applicant.

Furthermore, the conference bridge of Horn is said to include a "music detector," although it is not explained precisely how this function is accomplished. It is clear from the specification of Horn, however, that summing of an offending audio channel with other audio channels in a conference is suspended upon detection of "music-on-hold" signals. There is no teaching or suggestion that undesired audio message streams in general can be detected or removed by using Horn's technique.

Applicants' claims 29-32 depend ultimately from claim 1, which has been shown to be allowable above. In addition, contrary to the Examiner's assertion, neither Chan et al. nor Horn (as discussed above) deals with an intelligent network node or its functionality. Consequently, the subject matter of claims 29-32 cannot be said to be rendered obvious through combination of these references, and claims 29-32 are in condition for allowance for these additional reasons.

7. The Applicants respectfully submit that claims 1, 4-14, 23 and 25-34 have been shown to be patentably distinguishable over the prior art of record, and are thus in condition for allowance. Allowance of all claims pending is respectfully requested. If a telephone conference would be of assistance in advancing the prosecution of this application, the Examiner is invited to call applicants' attorney.

Respectfully submitted,

John W. Hayes

John W. Hayes
Attorney for Applicant
Reg. No. 33,900

Dated: August 16, 2005

PATTI & BRILL, LLC
Customer Number 32205